In the Specification:

Please replace the paragraph on page 1, lines 9-11, as follows:

Fields of the invention ~~includes~~ include noise cancellation. The invention concerns other more particular fields, including, but not limited to, active noise control using a feedforward or a feedback controller.

Please replace the paragraph on page 2, lines 21 through page 3, line 2, as follows:

One embodiment of the invention uses broadband feedforward sound compensation, which is a sound reduction technique where a sound disturbance is measured at an upstream location of the (noisy) sound propagation and cancelled at a downstream direction of the (noisy) sound propagation. An active noise control algorithm is the actual computation of a control signal (or compensation signal) that is able to reduce the effect of an undesired sound source by generating an out-of-phase sound source. To achieve proper sound cancellation, the active noise control algorithm must take into account the dynamic effects of the propagation of both the undesired and the out-of-phase sound source. The invention provides such a feedforward noise control algorithm and method that ~~take~~ takes into account the dynamic effects of sound propagation.

Please replace the paragraphs on page 3, lines 3-23, as follows:

The inventive active noise control algorithm described in this invention uses a FIR (Finite Impulse Response) filter where the orthogonal basis functions in the filter are chosen on the basis of the dynamics of the sound propagation. In this approach the standard tapped delay line of the FIR filter is replaced by a FIR filter that contains information on how the sound propagates through the system. The so-called generalized FIR (GFIR) filter has a much larger dynamic range while maintaining the linear parameter dependency found in a conventional FIR filter. As a result, adaptive and recursive estimation techniques can be used

to estimate the parameters of the GFIR filter. The GFIR filter requires an initialization that contains knowledge on sound propagation dynamics. Once actuators and sensors for active noise control have been placed in the system~~. The~~, the data from the actuators and sensors can be used to measure and characterize the dynamics of the sound propagation, and this information is used to initialize the GFIR filter.

Another embodiment of the invention concerns a feedback sound compensation system that treats the ~~affects~~ effects of both the periodic and non-periodic noise components. With the present invention, we are able to design a sound control algorithm that emphasizes the elimination of periodic components without over amplifying the non-periodic sound components. The controller is tuned to reject the periodic disturbances until there is no appreciable difference between the periodic and non-periodic disturbances.

Please replace the paragraph beginning on page 5, line 1 through page 6, line 22, as follows:

In order to analyze the design of the feedforward compensator 16, consider the block diagram depicted in FIG. 2. Following this block diagram, the dynamical relationships ~~relationship~~ between signals in the ANC system 10 are characterized by discrete time transfer functions, with $qu(t) = u(t+1)$ indicating a unit step time delay. The spectrum of noise disturbance $u(t)$ at the input microphone 12 is characterized by filtered white noise signal $n(t)$ where $W(q)$ 22 is a (unknown) stable and stable invertible noise filter. The dynamic relationship between the input $u(t)$ and the error $e(t)$ microphone signals is characterized by $H(q)$ 24 whereas $G(q)$ 26 characterizes the relationship between control speaker signal and error $e(t)$ microphone signal. Finally, $G_c(q)$ 28 is used to indicate the acoustic coupling from control speaker 18 signal back to the input $u(t)$ microphone 12 signal that creates a positive feedback loop with the feedforward $F(q)$. For the analysis, we assume in this that all transfer functions in FIG. 2 are stable and known. The error microphone signal $e(t)$ can be described by

$$e(t) = W(q)\left[H(q) + \frac{G(q)F(q)}{1 - G_c(q)F(q)}\right]n(t) \tag{1}$$

and is a stable transfer function if the positive feedback connection of *F(q)* 30 and $G_c(q)$ 28 is stable. When the transfer functions in FIG. 2 are known, perfect feedforward noise cancellation can be obtained in case

$$\begin{aligned} F(q) &= -\frac{H(q)}{G(q) - H(q)G_c(q)} \\ &= \frac{\tilde{F}(q)}{1 + \tilde{F}(q)G_c(q)}, \ \tilde{F}(q) := -\frac{H(q)}{G(q)} \end{aligned} \tag{2}$$

and can be implemented as a feedforward compensator 16 in case *F(q)* 30 is a stable and causal transfer function. The expression in equation (2) can be simplified for the situation where the effect of acoustic coupling G_c can be neglected. In that case, the feedforward compensator 16 can be approximated by

$$F(q) \approx \tilde{F}(q) = -\frac{H(q)}{G(q)} \tag{3}$$

and for implementation purposes it would be required that *F(q)* 30 be a causal and stable filter. In general, the filter *F(q)* 30 in equation (2) or (3) is not a causal or stable filter due to the dynamics of *G(q)* 26 and *H(q)* 24 that dictate the solution of the feedforward compensator. Therefore, an optimal approximation has to be made to find the best causal and stable feedforward compensator. With equation (1) the variance of the discrete time error signal *e(t)* is given by

$$\frac{\lambda}{2\pi}\int_{-\pi}^{\pi} |W(e^{j\omega})|^2 \left| H(e^{j\omega}) + \frac{G(e^{j\omega})F(e^{j\omega})}{1 - G_c(e^{j\omega})F(e^{j\omega})} \right|^2 d\omega$$

where λ denotes the variance of *n(t)*. In case variance minimization of the error microphone signal *e(t)* is required for ANC, the optimal feedforward controller *(F)* 16 is found by the minimization

$$\begin{aligned}
&\min_{\theta} \int_{\omega=-\pi}^{\omega=\pi} |L(\mathrm{e}^{j\omega},\theta)|^2 \mathrm{d}\omega := \min_{\theta} \|L(q,\theta)\|_2, \\
&L(q,\theta) = W(q)\left[H(q) + \frac{G(q)F(q,\theta)}{1 - G_c(q)F(q,\theta)}\right]
\end{aligned} \tag{4}$$

where the parametrized filter $F(q,\theta)$ is required to be a causal and stable filter, in which θ is a real valued parameter determined by the minimization in equation (4).